Watermarking Algorithm based on Spread Spectrum Technique

A Report

Submitted in partial fulfilment of the requirements for the award of the Degree of

Bachelor of Technology

in

Computer Science and Engineering

By

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ABSTRACT

Abstract—In the realm of digital audio watermarking, the embedding of significant watermark signals into original audio signals is pivotal for subsequent extraction as needed. This technology holds immense value in domains such as copyright protection, content authentication, secure communication, and anti-counterfeiting endeavors. With the widespread availability of multimedia software and audio editing tools, the ease of audio modification has increased significantly. However, a majority of existing digital audio watermarking algorithms fall short in their ability to withstand commonplace operations such as shifting, editing, tone manipulation, and splicing. To address this limitation, this paper presents a novel audio watermarking algorithm explicitly designed to withstand variable speed attacks on audio signals. Leveraging the WSOLA (Waveform Similarity Overlap and Add) principle, the proposed algorithm intelligently embeds a spread spectrum watermark within non-overlapping regions during the audio speed change process. Extensive experimentation demonstrates the algorithm's heightened robustness and imperceptibility, making it highly suitable for practical applications.

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Date: 15-07-2023 Sameer Damkondwar

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CHAPTER 1 : INTRODUCTION

In today's digital age, the rapid growth of big data and the widespread availability of music streaming platforms like QQ Music and NetEase Cloud Music, along with various audio editing software such as Adobe Audition, have made it incredibly convenient and easy for people to access and manipulate digital media. However, this convenience has also given rise to significant challenges in terms of copyright infringement and piracy, posing a serious threat to the rights of copyright owners. To combat these issues, there is an urgent need for effective methods to identify and protect digital media content from unauthorized use. Digital watermarking technology has emerged as a promising solution for copyright tracking and verification. By imperceptibly embedding special watermark information into digital media, it allows for the extraction of the watermark when copyright disputes arise, enabling the tracking and validation of copyrights.

Digital watermarking technology encompasses various forms, including text, image, audio, and video watermarking. In the context of audio watermarking, there are two main processes: embedding and extracting. The goal is to invisibly embed watermark information of significant importance into digital audio without compromising its original quality and usability. This watermark remains embedded within the audio as an integral part of the data, and can be extracted when necessary to verify the copyright.

However, the challenges of copyright protection extend beyond embedding watermarks. During the transmission and usage of audio, various attacks can compromise the integrity of the watermark. If the watermark information can be easily destroyed or altered, the purpose of copyright protection is defeated. Therefore, it is essential to develop audio watermarking algorithms that are robust against attacks while ensuring that the watermark remains imperceptible. Particularly, the robustness of existing audio watermarking algorithms against variable speed attacks needs improvement.

To illustrate this, let us consider an example: Imagine a popular song that has been digitally watermarked to protect its copyright. Now, suppose someone decides to use this song in their own project but wants to modify it by increasing or decreasing its playback speed. During the speed change process, the watermark embedded in the original song may become distorted or even lost entirely. As a result, it becomes impossible to accurately extract the watermark and validate the copyright, leading to potential misuse and infringement.

To address this problem, this paper proposes a novel audio watermarking algorithm that aims to enhance the robustness of watermarks against variable speed attacks. By leveraging innovative techniques based on the principles of audio processing, the algorithm intelligently embeds the spread spectrum watermark into non-overlapping regions when the audio speed is modified. This ensures that the watermark remains intact and extractable, even in the presence of variable speed changes.

While several audio watermarking algorithms have been proposed, they often struggle to preserve the integrity of the watermark when facing speed changes. Through a thorough analysis of existing techniques and considering their limitations, this paper introduces a new and improved audio watermarking algorithm. This algorithm prioritizes robustness against variable speed attacks while maintaining imperceptibility. Experimental evaluations will be conducted to validate the algorithm's effectiveness and compare its performance with existing methods.

In summary, this paper addresses the critical need for robust audio watermarking algorithms to combat copyright infringement in the digital age. By proposing an innovative approach specifically designed to withstand variable speed attacks, the algorithm aims to enhance the success rate of copyright protection while ensuring the watermark remains imperceptible. The subsequent sections will provide a detailed explanation of the proposed algorithm, experimental results, and comparisons with existing methods, ultimately demonstrating its effectiveness and practicality in real-world scenarios.

CHAPTER 2: LITERATURE REVIEW

Audio watermarking algorithms have undergone significant advancements in recent years to address the challenges of imperceptibility, robustness, and security in the field. Various techniques have been proposed and explored, aiming to embed imperceptible and resilient watermarks within audio signals for applications such as copyright protection and content authentication.

Spread spectrum-based watermarking techniques have been widely investigated. These methods spread the watermark signal across the audio spectrum, making it resistant to signal processing attacks. By leveraging properties of the human auditory system, echo hiding techniques exploit auditory masking effects to embed watermarks imperceptibly. Quantization-based methods have also gained attention, employing techniques like discrete cosine transform (DCT) to embed watermarks while maintaining high audio quality.

Transform domain techniques, including discrete Fourier transform (DFT), DCT, and wavelet transform, have been extensively studied. These methods allow for the localization and manipulation of audio features. They have been used in watermarking algorithms that combine frequency and time-frequency domains to embed watermarks, achieving robustness against various attacks while preserving perceptual quality.

While numerous audio watermarking algorithms have been proposed, challenges such as synchronization, robustness against advanced attacks, and adaptability to different audio formats remain open research areas. Future efforts should focus on addressing these challenges and developing efficient and secure audio watermarking algorithms suitable for emerging applications, such as real-time streaming and cloud-based audio services.

In conclusion, this literature review provides an overview of the advancements in audio watermarking algorithms, including spread spectrum, echo hiding, quantization-based methods, and transform domain techniques. These techniques offer different trade-offs in terms of imperceptibility and robustness. However, further research is needed to overcome existing challenges and enhance the performance of audio watermarking algorithms in real-world scenarios.

CHAPTER 3: METHODOLOGY

WSOLA Algorithm

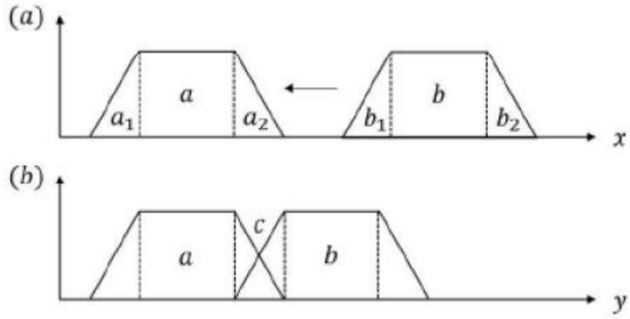

Figure 1 Extracting Process

As shown *figure (a)* x in the figure is the audio input without speed change, *figure (b)* y in the figure is the audio output after x speed change. First, process the first frame, copy the data of area a to output y, and then a_2 , and b_1 , data is superimposed, the superimposed result C is copied to output y , and finally the data of area b is copied to output y, that is, the processing of one frame is completed. The algorithm in this chapter selects the non-overlapping regions in each frame for operation, such as region a and region b.

A. Embedding Algorithm

In the embedding phase, we have tempo (speed parameter), overlapLength (length of overlapping area), seekLength (length of the offset search range), and seekwindowLength (sum of frame data length and overlap length). We also use nominalskip, which determines the step size for frame processing based on formula:

$$\textbf{\textit{skipFract = skipFract + nominalskip}}$$

$$\textbf{\textit{nomSkip= tempo × (seekwindows – overlap)}}$$

Here’s the embedding algorithm for embedding a given watermark in the audio sample imperceptibly:

1. For randomly generated seed sequence b0, cyclic shift, generating the matrix $b_0, b_1, \ldots b_{l-1}$, by performing a cyclic shift operation on the seed sequence. This creates a matrix of shifted sequences, where l represents the length of the seed sequence.

2. Perform Schmidt orthogonalization on the matrix to obtain an orthonormal matrix P. Schmidt orthogonalization is a process that transforms a set of vectors into an orthonormal set by iteratively subtracting the projections of the vectors onto the previously computed orthogonal vectors.

3. Select 2^b sequences from matrix P to create a pseudo-random sequence matrix. These selected sequences, denoted as $p_1,p_2,...p_{2^b}$, will be used for watermark embedding.

4. Segment the audio signal into frames of equal size. Each frame represents a portion of the audio signal.

5. Apply the Discrete Cosine Transform (DCT) to each audio frame. The DCT transforms the time-domain audio signal into the frequency domain.

6. For each frame, perform the following steps, to implement the embedding formula, to embed the watermark information:

$$\boldsymbol{y_i = x_i + w_i \,.\, sign(x_i \,.\, p_t^T)p_t}$$

a) Obtain the DCT coefficients of the frame, denoted as $x_1,x_2,...x_N$.
b) Retrieve the corresponding pseudo-random sequence from $p_1,p_2,...p_{2^b}$, based on the frame index.
c) Calculate the watermark bit w_i to be embedded, based on the watermark information.
d) Compute the inner product of x_i and the selected pseudo-random sequence p_t.
e) Multiply the inner product by w_i and take the sign value.
f) Modify the DCT coefficient x_i by adding the product of w_i, the sign value, and the selected pseudo-random sequence p_t.
g) Repeat steps a to f for each frame.

7. After embedding the watermark into all the audio frames, perform the inverse DCT (IDCT) transform on each frame to obtain the modified audio segments.

8. Reconstruct the watermarked audio signal by combining the modified audio segments.

The above algorithm describes the embedding process of the audio watermarking technique. It uses a combination of DCT, pseudo-random sequences, and embedding formulas to modify the audio frames based on the provided watermark information.

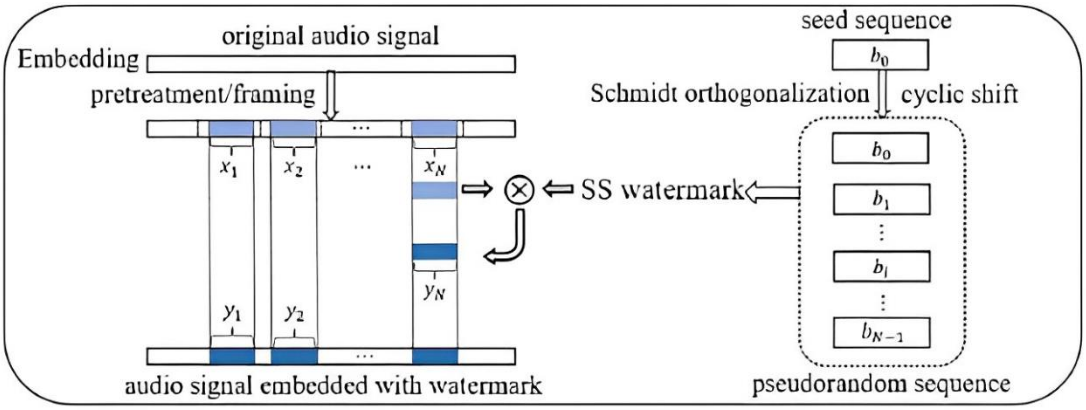

Figure 2.1 Embedding Process

B. Extracting Algorithm

This algorithm describes the process of extracting a watermark from an audio signal using a watermark embedding method based on the WSOLA (Waveform Similarity Overlap and Add) algorithm. Here's a step-by-step explanation of the algorithm along with the given formula:

1. The algorithm begins by determining the position of the watermark in each frame of the audio signal. This is done using the known values: overlapLength, seekwindowLength, seekLength, and the offset value of each frame. These parameters help identify the specific region where the watermark is embedded.

2. Next, the algorithm performs Schmidt orthogonalization on a matrix P. This process involves generating a sequence of pseudo-random watermark sequences $p_1, p_2, \ldots p_2^b$. To obtain these sequences, a randomly generated seed sequence b0 is cyclically shifted to generate $b_0, b_1, \ldots b_{l-1}$. From this matrix, 2^b sequences are selected to be used as the pseudo-random watermark sequence matrix.

3. In this step, the algorithm applies the Discrete Cosine Transform (DCT) to the area of interest

in each frame. The DCT transformation converts the audio samples into a set of DCT coefficients, represented as vector $x_1, x_2, \ldots x_N$.

4. Finally, the watermark is extracted from each frame using the extraction formula provided. For each frame, the algorithm computes the correlation between the pseudo-random watermark sequence vectors and the transformed audio vectors. The correlation is calculated using the formula:

$$t = \underset{j \in \{1,2,\ldots,2^b\}}{\arg\max} R_j = \underset{j \in \{1,2,\ldots,2^b\}}{\arg\max} \left| y p_j^T \right|$$

In this formula, j ranges from 1 to 2^b, representing the indices of the pseudo-random watermark sequences. The variable t corresponds to the extracted watermark sequence, which is obtained by converting the index j into binary.

By performing these steps on each frame of the audio signal, the algorithm extracts the watermark embedded in the audio. The watermark extraction process is the reverse of the watermark embedding process, allowing the original watermark information to be retrieved from the watermarked audio.

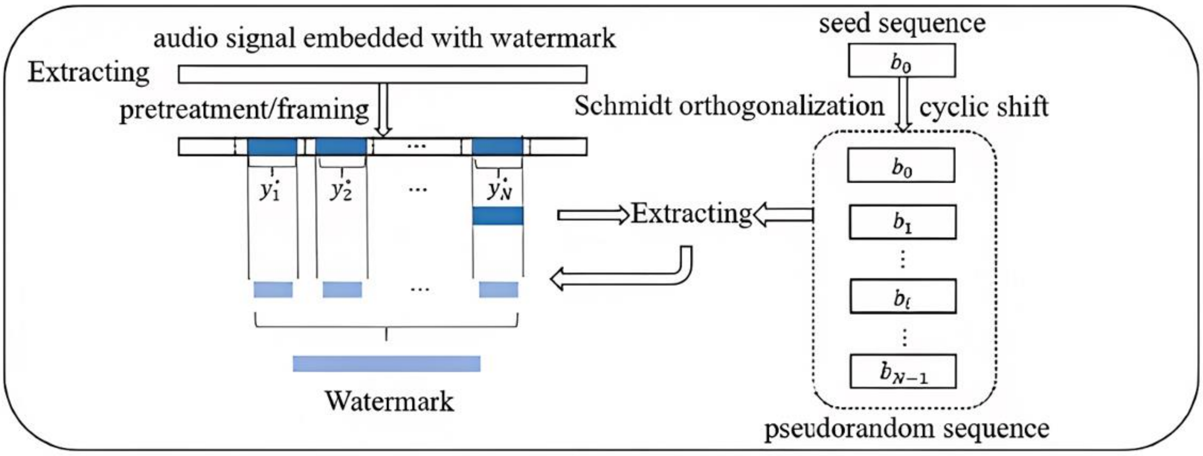

Figure 2.2 Extracting Process

CHAPTER 4: EXPERIMENTAL RESULTS AND DISCUSSION

Imperceptibility refers to the absence of noticeable distinctions between the original audio, free of any watermark, and the audio containing an embedded watermark. In other words, the process of embedding an audio watermark should not affect the quality or value of the audio content. The Signal-to-Noise Ratio (SNR) serves as the benchmark for evaluating the imperceptibility.

The SNR (Signal-to-Noise Ratio) measures the level of audio distortion that occurs when a watermark is embedded. A higher SNR indicates lower audio distortion, meaning that the watermark is more imperceptible. In the formula, $\bar{x}$ represents the original audio signal, and $\hat{x}$ represents the audio signal with the embedded watermark.

$$SNR = 10\log_{10}\left(\frac{\sum (x)^2}{\sum (noise)^2}\right)$$

$$MSE = \sum\left(\frac{(x-y)^2}{length(x)}\right)$$

$$PSNR = 10\log_{10}\left(\frac{(peakSignal)^2}{MSE}\right)$$

$$NRMSE = \sqrt{\left(\frac{\sum (x-y)^2}{\sum (x)^2}\right)}$$

x = original signal

y = noisy signal

noise = difference between the noisy signal and the original signal (noise = y - x)

peakSignal = maximum possible signal value in the original signal

MSE = Mean Squared Error between the original signal and the noisy signal

PSNR = Peak Signal-to-Noise Ratio

NRMSE = Normalized Root Mean Square Error

Experimental environment:

11th Gen Intel Core i5, 2.40 GH z - x64 processor, 8 GB RAM, 64-bit operating system

Algorithm Language:

MATLAB

Experimental Audio:

We used multiple WAV file viz. *sample1.wav, sample2.wav, sample3.wav, sample4.wav* & *sample5.wav* for the experiment.

Experiment Results:

We performed a speed change test on the "sing.wav" file and obtained the results presented in Table 1. Based on our findings, we can conclude that the algorithm demonstrated robustness against variable speed attacks.

We performed the embedding of a watermark on five different audio samples listed above. The findings of the embeddings are provided in the table below:

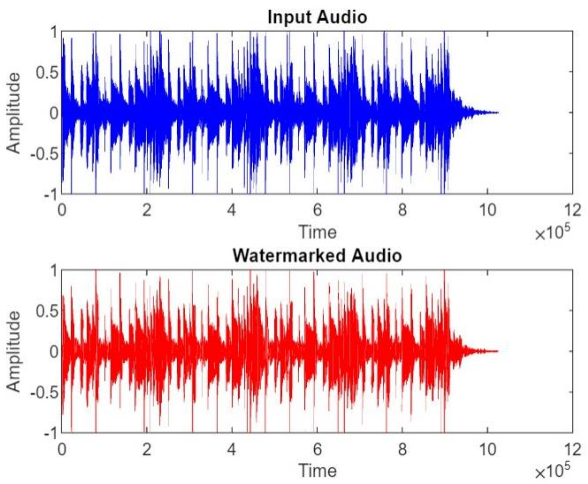

Figure 3.1 Sample 1

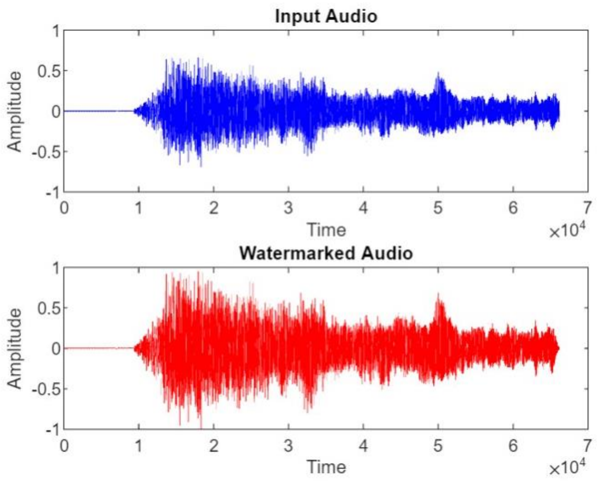

Figure 3.2 Sample 2

Algorithm	Audio	SNR	PSNR	NRMSE
DCT	sample1.wav	40.20 dB	13.99 dB	0.34
	sample2.wav	5.15 dB	28.50 dB	0.33
	sample3.wav	46.30 dB	15.63 dB	0.98
	sample4.wav	6.21 dB	15.23 dB	0.63
	sample5.wav	2.56 dB	12.14 dB	0.52

Table 1 Experimental Results on Audio Quality after embedding

CHAPTER 5: CONCLUSION

In conclusion, this paper introduces a novel audio watermarking algorithm that leverages the WSOLA (Waveform Similarity based Overlap-Add) algorithm for embedding watermarks in audio signals. The algorithm incorporates a spread spectrum-based pseudo-random sequence to enhance the randomness and robustness of the watermark.

By utilizing the WSOLA algorithm, which is known for its ability to analyze waveform similarity and perform overlap-add operations, the proposed algorithm achieves effective watermark embedding and extraction without compromising the integrity of the audio content. The WSOLA algorithm provides valuable insights into the relationship between audio signals, allowing for the identification of invariants crucial for successful watermarking.

The integration of a spread spectrum-based pseudo-random sequence enhances the security of the watermark, ensuring its resilience against potential attacks or removal attempts. This approach introduces randomness to the watermark, making it more difficult for unauthorized users to detect or remove.

The presented algorithm showcases promising potential for various applications, such as copyright protection and intellectual property identification in audio content. It contributes to the field of audio watermarking by offering an innovative solution that combines the power of the WSOLA algorithm and spread spectrum techniques to achieve secure watermarking capabilities.

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